


## Paper Type: Original Article



# QoS-Aware Enhanced Proportional Fair Scheduling Algorithm for Real-Time Services in LTE Networks

Mahnaz Sotoudeh Bahreyni<sup>1</sup> , Vahid Sattari-Naeini<sup>2</sup><sup>1</sup> Department of Information Technology Engineering, Science and Research Branch, Islamic Azad University, Kerman, Iran; mahnaz.sotoodeh@gmail.com.<sup>2</sup> Department of Computer Engineering, Shahid Bahonar University of Kerman, Kerman, Iran; vsattari@uk.ac.ir.

## Citation:

Sotoudeh Bahreyni, M., & Sattari-Naeini, V. (2023). QoS-aware enhanced proportional fair scheduling algorithm for real-time services in LTE networks. *Computational algorithms and numerical dimensions*, 2(2), 102-112.

Received: 15/01/2023

Reviewed: 16/02/2023

Revised: 09/04/2023

Accepted: 10/05/2023

## Abstract

The new generation of wireless networks (LTE advance and WIMAX) supports many services that consume many resources (such as VOIP, video conference ...). Adding multi-media services to wireless communication systems provide new challenges of resource allocation. This paper proposes a resource scheduling downlink algorithm for LTE networks. In the proposed algorithm for different types of services, priorities are defined to guarantee transitions of GBR services that need high QoS. This method also considers channel quality and buffer status to achieve higher throughput for non-GBR services. The proposed algorithm is simulated and compared with the proportional fair algorithm. Simulation results show that the suggested algorithm can increase system throughput and QoS of real-time services at the cost of a certain amount of throughput and QoS of non-real times.

**Keywords:** LTE networks, Scheduling, GBR services, Non-GBR services, Quality of service, Downlink.

## 1 | Introduction



Computational  
Algorithms and  
Numerical Dimensions.

This article is an open access article distributed under the terms and conditions of the Creative Commons Attribution (CC BY) license (<http://creativecommons.org/licenses/by/4.0>).

LTE is a new standard specified by 3GPP and is moving towards fourth-generation wireless networks. In recent years, the number of internet users who are accustomed to high-speed internet access (wherever they go, not just the home or office) are increasing therefore, mobile high bandwidth has become a reality. LTE is a response to moving users that consume a lot of bandwidth, to provide them better services and applications. LTE introduces mechanisms for supporting the implementation of new data services in the network by increasing the data rate. LTE is included a few nodes in the network core, then it reduces the protocol processing overload and leads to reduced delay [1]-[4]. Scheduling is the process of deciding by the scheduler for the allocation of resources (time, frequency) between users in a communication system. In LTE networks, scheduling the resources is very important because a good performance can be achieved by properly allocating resources to the users. Different methods are suggested to use radio resources effectively and to meet the quality of service requirements for different types of traffic. In [5] and [6] performance of different scheduling algorithms in LTE is evaluated based on various parameters. In [7] a time and frequency domain scheduling scheme is proposed which is implemented in two layers. In the first



Corresponding Author: mahnaz.sotoodeh@gmail.com

<https://doi.org/10.22105/cand.2023.432830.1083>

layer, an algorithm has been implemented in the time domain to maintain fairness in the allocation of radio resources among users. Another scheduling algorithm is implemented in the frequency domain to improve throughput. In general, these algorithms focus on increasing throughput and maintaining equal fairness between users, but do not consider the restriction of delayed multimedia traffic. Also, this schema doesn't take into account user guaranteed bit-rate requirements. A scheduling technique for cellular networks is suggested in [8]. The proposed scheme has two parts: 1) "scheduler", and 2) "resource allocation". The first part, "scheduler", selects traffic flows and determines the amount of allocated bandwidth for each traffic flow. The second part, "resource allocation", determines which sub-channels are allocated to each traffic flow. In [9] a new scheduling algorithm is presented. The proposed algorithm makes a balance between spectral efficiency (in terms of throughput) and fairness, also it can work on the cell edge while preparing appropriate fairness. In [10] a modified PF scheduling algorithm is proposed and compared with the Proportional Fair (PF) algorithm. The results show that the algorithm can also increase the average cell throughput and distribute resources fairly. These proposed algorithms are not controlled the real-time traffic flows effectively. In [11] a scheduling mechanism for real-time traffic flow is proposed. The impact of different packages in the queue for each user, based on different evaluation parameters is compared and analyzed. This mechanism has been used only in the case of video traffic flows. In [12], a QoS aware scheduling strategy is used and the load cell is investigated. Two new scheduling strategies based on the PF scheduling algorithm are proposed and compared with the performance of classical scheduling strategies, in terms of delay and throughput. The algorithms have been used in different traffic situations (deferent services under deferent network load). In [13] a scheduling scheme has been suggested that makes difference between real-time and non-real-time connections. Also, it considers problems such as starvation and fairness. To meet the requirements of service quality, ensure that no connection in the network is not occupied recourses for long periods or connections that are waiting for a long time should be served. Ma et al. [14] and Asadollahi and Dehdasht-Heydari [15] proposed a QoS guaranteed scheduling algorithm for multimedia systems in wireless networks. In this approach, to ensure the quality of service with higher priority users, the performance of lower priority users, such as NRT streams, is reduced. Costa Neto et al. [16] and Chaudhuri et al. [17] proposed a QoS-aware scheduling algorithm to guarantee the user's quality of service in OFDMA systems. Algorithms are designed in both uplink and downlink directions. Since in these approaches, the type of user service is not considered, the total quality of the service system is reduced. In [18] outers introduce two QoS-aware algorithms for edge users in LTE networks. One of the algorithms is in terms of fairness parameter, and the other is in terms of fairness and bandwidth parameters; which finally increase the fairness and throughput of edge users at the network. The performance of these algorithms is compared with the performance of existing algorithms. Madi et al. [19] and Rocha and Vieira [20], have investigated the effect of latency on improving the quality of service for real-time streams in different speed scenarios. In these approaches, less importance is given to other service quality parameters as well as guaranteeing the minimum quality of service for non-real-time traffic. In [21] a scheduling algorithm for delay-sensitive applications on LTE networks is proposed. In this approach, to achieve a level of service quality, a delay threshold is considered, and in allocating free resource blocks to users, features related to the data rate average and the number of missing packets are considered. In the proposed strategy, with reeducating the number of missing packets, the throughput of the whole system has increased. Paper [22] are suggested a hybrid QoS-aware scheduling algorithm that considers both delay and queue parameters in its scheduling metric. This approach by balancing QoS for various kinds of traffics provides better total system throughput. In [23] a comparative study has been performed between BCQI, PF, RR, FFR algorithms in the next-generation wireless communications, and the performance of these algorithms is evaluated in terms of throughput, fairness, performance, and throughput of the whole system.

The proposed algorithm in this paper prioritizes users based on the buffer status and packet specifications. This algorithm makes a maximum throughput system and ensures that delay for a user is never more than the delay threshold. Also, a user receives a minimum throughput until its quality of service requirements are fulfilled. QoS requirements have been met by allocating more resources to those users who have critical delay and throughput (high latency and low throughput). The proposed algorithm with prioritization of services and taking into account the user buffer status minimizes delay and packet loss rate. Initially, in Section 2, scheduling algorithms in LTE are studied. In Section 3, the proposed algorithm is presented.

The simulation parameters are given in Section 4. The simulation results are presented in Section 5. Finally, in Section 6 the results of the study are presented.



## 2 | Scheduling Algorithms for LTE Networks

### 2.1 | Round Robin Scheduling Algorithm

Round Robin (RR) scheduling algorithm is an unaware scheduling scheme that allows users to use shared resources in turns, without taking into account their channel conditions. So, this radio resource allocation method makes the best fairness among users. But, system throughput performance will decrease.

### 2.2 | Best CQI Scheduling Algorithm

Best CQI scheduling algorithm assigns radio resources to the user's most received CQI. This scheduling algorithm with assigning resources to the good channel quality user helps to improve throughput users; thus causes increases the system data rate. This algorithm increases system throughput in the cost of decreased fairness.

### 2.3 | Proportional Fair Scheduling Algorithm

This algorithm allocates more resources to the user who has relatively better channel quality; the combination of the CQI and the level of fairness. So, the highest throughput cell with a level of fairness will provide. This algorithm's main purpose is to achieve a balance between the maximum throughput and fairness.

## 3 | Proposed Scheduling Algorithm

In the proposed algorithm, guaranteed bit rate service has priority to the non-guaranteed bit rate service, and allocation of resources to the not-guaranteed bit rate service is performed after guaranteed bit rate service. When there are a lot of users with bad channel quality on the system, a large number of resources are occupied for the transmission of guaranteed bit rate data. This leads to a reduction in the throughput system. To avoid such a situation, a threshold ( $\alpha$ ) is defined. ( $\alpha$ ) is a percentage of resources that guaranteed bit rate services can be used and indicates permission of guaranteed bit rate services for using resources.  $\alpha$  is an experimental value and it is changeable and can be set based on the number of users and the percentage of guaranteed bit rate services in the system. In the case of, there is no guaranteed bit rate source while there is still guaranteed bit rate service, Service to users will end. If all guaranteed bit rate services are serviced before the guaranteed bit rate resources run out, the remaining resources will be allocated to the non-guaranteed bit rate services. The proposed scheduling algorithm will be explained in two stages: 1) resource allocation to guaranteed bit rate service, and 2) resource allocation to non-guaranteed bit rate service.

### 3.1 | Resource Allocation to Guaranteed Bit Rate Service

We must also consider channel conditions and packet delay during resource allocation to the guaranteed bit rate services. The formula for calculating the priority of guaranteed bit rate services is given below (Eq. (1)). Using the PF algorithm in [2], eNodeB allocates the number of N resource blocks to the K\* set users, during t TTI.



$$K^*(t) = \arg \max \frac{[R_K(t)]^\alpha}{[T_K(t)]^\beta}. \quad (1)$$

That  $R_k(t)$ ,  $k = 1, 2, \dots, z$  is the instantaneous bit rate for  $k$ th users (in  $t$ th TTI).  $T_k(t)$  is also the average throughput of the user, which is exponentially updated using a low-pass weighting filter.

$$T_k(t+1) = \begin{cases} \left(1 - \frac{1}{\tau_c}\right) T_k(t) + \frac{1}{\tau_c} R_k(t), & k \in k^*, \\ \left(1 - \frac{1}{\tau_c}\right) T_k(t), & k \in k^*. \end{cases} \quad (2)$$

$\tau_c$  is the window size. In Eq. (1) PF algorithm with  $(\alpha=1, \beta=1)$  makes a balance between throughput and fairness. Other values of  $\alpha$  and  $\beta$  can be used to achieve the desired balance between throughput and fairness. So that, we can calculate the priority equation of the user's  $k$  as follows:

$$P_K(t) = \frac{[R_K(t)]^\alpha}{[T_K(t)]^\beta}, \quad \beta = \alpha = 1. \quad (3)$$

Buffer overflows result in large transfers, leading to packet loss. In the proposed algorithm user buffer status is considered to prevent packet loss due to overflows. Buffer status is obtained by sending information of the user buffer in the uplink direction. This information includes the length of receiver buffer ( $L_{buffi}$ ) and length of buffer used space ( $C_{buffi}$ ), which are forwarded by user  $i$  and can be calculated as follows [24]:

$$E_{buffi} = L_{buffi} - C_{buffi}. \quad (4)$$

$$buff_i = \frac{L_{buffi} - E_{buffi}}{L_{buffi}}. \quad (5)$$

In Eq. (5),  $E_{buffi}$  is the buffer free-space length of the user,  $buff_i$  is the weighting factor of the user buffer. The user with greater factor has fewer buffers free space, therefore will give greater priority to earlier scheduling, and vice versa. According to [25] delay priority factor is defined as follows:

$$D_k(t) = \sum_{j=1}^M \gamma_j \frac{D_{j,k}(t)}{D_{j,constr}}. \quad (6)$$

$\gamma_j$  is the weighting factor for traffic  $j$  (there are  $M$  kinds of traffic),  $D_{j,k}(t)$  is the estimated delay for the user  $k$  at the traffic  $j$ ,  $D_{j,constr}$  is the maximum delay allowed for the user  $k$  in traffic  $j$ ,  $\gamma_j$  is weighting factor for traffic class  $j$  such that

$$\sum_{j=1}^M \gamma_j = 1. \quad (7)$$

In Eq. (8),  $U_k(t)$  is user-factor  $k$  which is built based on the optimization Lagrange Multiplier (LM) method [26], [27].  $U_k(t)$  is defined as follows:

$$U_k(t) = \max \left( 0, \frac{r_{\sum constr} - r(t-1)}{\tau_c} \right). \quad (8)$$

Where  $r(t-1)$  is the data rate of user  $k$  in TTI  $(t-1)$ ,  $r_{\sum constr} = \sum_{j=1}^M r_{j,constr}$  is the sum of the minimum acceptable data rate for the user  $k$ ,  $r_{j,constr}$  is the minimum acceptable data rate for traffic  $j$  based on the ITU-T Rec. G.1010. From Eqs. (3), (5), (6), and (8) the priority function of the proposed algorithm for user  $k$  is as follows:

$$P_k(t) = \frac{R_k(t)}{T_k(t)} \times U_k(t) \times D_k(t) \times buff_i. \quad (9)$$

Then packets are sorted based on priority calculated in Eq. (9) by descending order. First, the package with the highest priority, then the other packages with the lower priority will be scheduled. In the proposed algorithm, after allocating resources to guaranteed bit rate users, the latency and data rates

of these users are measured to be ensured of good service quality. If the QoS of guaranteed bit rate users is not good, more resources (according to a pre-defined rate) will be allocated to them.

$$k^* (t) = \arg \max_{k=1,2,\dots,k}^N \frac{R_k(t)}{T_k(t)} \times U_k(t) \times D_k(t) \times \text{buff}_i. \quad (10)$$



### 3.2 | Resource Allocation to Non-Guaranteed Bit Rate Service

After resource allocation to the guaranteed bit rate service, resources are allocated to the non-guaranteed bit rate service. With the attitude of less need for QoS and also to reduce complexity, the PF algorithm is used as a reference, and also delay factor has been ignored. The calculation formula is as follows:

$$P_k (t) = \frac{R_k(t)}{T_k(t)} \times \text{buff}_i. \quad (11)$$

$R_k(t)$  is spectral efficiency user  $k$ ,  $T_k(t)$  is the history of throughput service  $i$ ,  $\text{buff}_i$  is the weighting factor of the user buffer. These factors can be calculated as before. To illustrate how the scheduling algorithm service and manage the mixed traffic of different services, users with five different services (VOIP, Video, Gaming, FTP, and HTTP) have been defined. Some services should be served with higher priority, because of their real-time nature. We'll investigate what would happen if all these services be considered in one class without discrimination between them. And, what would happen if we separate them. In this simulation,  $\alpha=0.6$  is considered. The flowchart of the suggested algorithm is shown in Fig. 1.

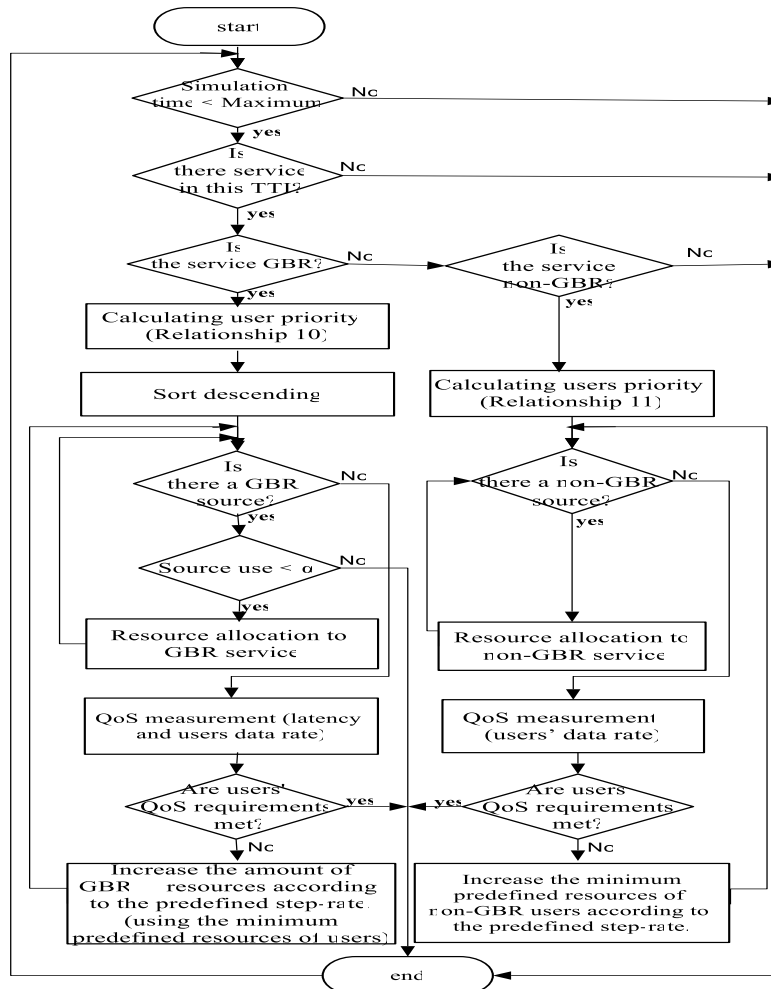


Fig. 1. Flowchart of the suggested algorithm.

## 4 | Simulation Parameters

Simulation is done in the frequency band GHz 2 and bandwidth 5 MHz. A total of 25 source blocks (5 kHz/ 180KHz = 25RBS) are available on the network and each resource block (RB) is composed of a band 180 kHz. Users are considered walking at a speed of 5 km/h. In addition, the fading model is used in order to simulate actual channel conditions. Other important parameters of the simulation are given in *Table 1*.

**Table 1. Simulation parameters.**

Parameter	Value
Frequency	GHz 2.0
Bandwidth	MHz 5
Antennas nTX and nRX	1rx,1tx
Transmission mode	SISO
Simulation length	TTI 100
Number of source blocks	25
Number of users	17, 11, 5, 24, 29, 33
Distance between eNodeBs	500M
eNodeB transmission power	20W
Number of simulations	200 in the scenario
Subcarriers in RB	12
Minimal connection loss	70Db

## 5 | Simulation Results

In this section we will evaluate the performance of the proposed algorithm with the PF algorithm which performs the scheduling process regardless of the type of traffic; also this algorithm is used a lot as a reference for comparison in the many papers. Then, these two algorithms will be compared together. Evaluating the performance of both scheduling algorithms is done with an LTE system level simulator [28]. Evaluation is done based on important parameters such as throughput system, Throughput Fairness Index, and packet delay. Before presenting the results, it is better to first define the performance of these evaluation parameters.

### 5.1 | Throughput Fairness Index

Fairness may be defined in terms of resource allocation or Throughput. To get the Throughput Fairness Index, Jain's equation will be used that has been defined as follows:

$$J(x_1, x_2, \dots, x_n) = \frac{(\sum_{i=1}^n x_i)^2}{n \sum_{i=1}^n x_i^2}. \quad (12)$$

In this equation,  $n$  is the number of users and  $x_i$  is the throughput for the  $j_{th}$  user. In the best case, its value is 1 which means that all users have gained the same Throughput. When the differences between users' throughputs increase, the value of this equation should be reduced [29].

### 5.2 | System Throughput (Downlink)

System Throughput is all users' data rate in bit/s.



### 5.3 | Packet Delay

Packet delay is the time difference between packet acknowledgment by the user and its creation, as follows [7]:

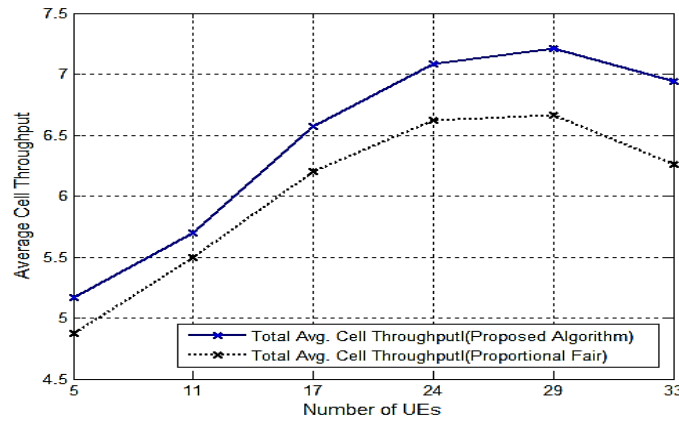
$$\text{Packet delay} = (\text{packet acknowledge time}) - (\text{Packet Generation time}). \quad (13)$$

To illustrate how the scheduler manages and services the different services, users with different QoS requirements (FTP, HTTP, Video, VOIP, and Gaming) are defined. As in Table 2, real-time services should be served with higher priority because of their real-time nature.

**Table 2. Services settings.**

Specifications QoS	QCI Degree	Service Type	Service Name
Time limit and minimum stored rate	1	GBR	VOIP
Maximum delay limit	2	GBR	Video
Maximum delay limit	4	GBR	Gaming
Minimum rate saved	8	Non-GBR	FTP
Required rate or no delay	8	Non-GBR	HTTP

Fig. 2 shows that the system throughput in the proposed algorithm is more than the PF algorithm. When the number of users in the system is low, all telecommunications services can be served, so the difference in throughput is low. As the number of users in the system increase, there will not be enough resources, so this difference will increase. The reason is that the PF algorithm does not consider the packet delay constraint. Packets are lost before they reach the user, so the average throughput dramatically decreases.



**Fig. 2. Comparison of the cell throughput.**

Figs. 3-5 respectively show that the throughput of VOIP, Video, and Gaming on the proposed algorithm is more than PF algorithm and this difference increases with increasing users. This is because the priority is given to this type of service, so the resources will be allocated to them first. It shows that the proposed algorithm can increase guaranteed bit rate carriers' throughput.

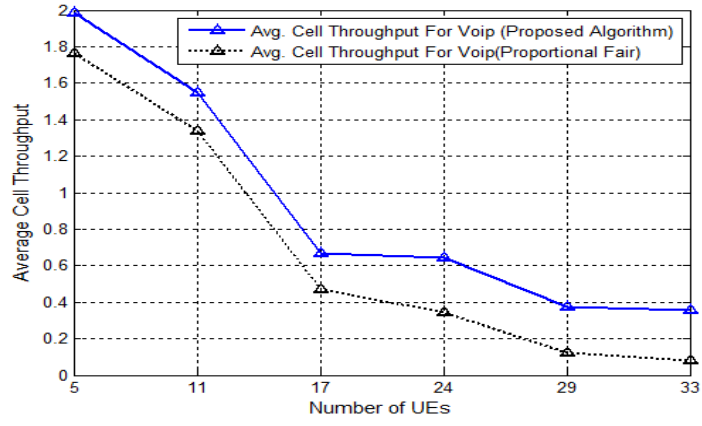


Fig. 3. Comparison of the VOIP throughput.

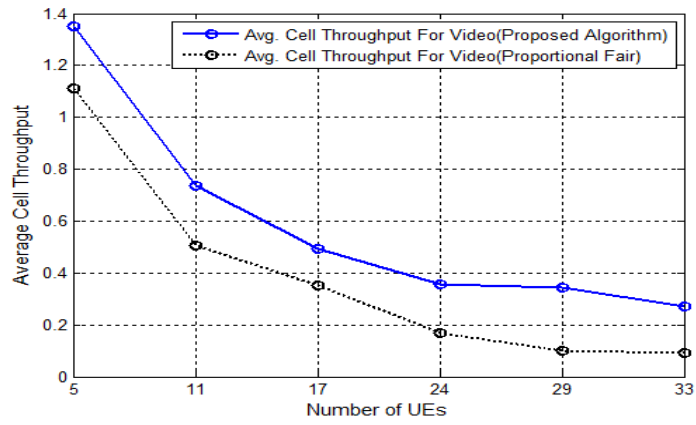


Fig. 4. Comparison of the Video throughput.

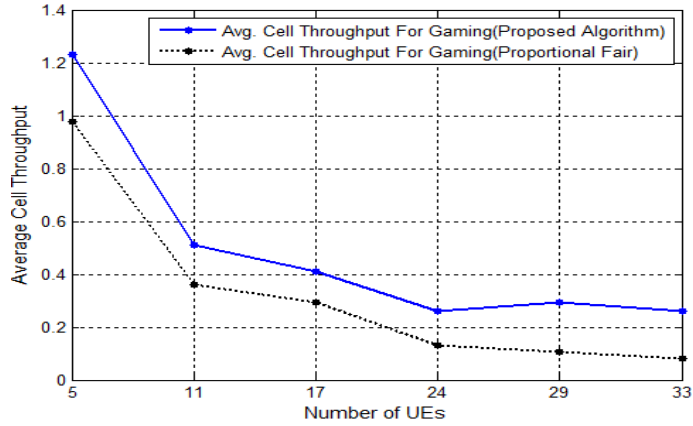


Fig. 5. Comparison of the Gaming throughput.

Fig. 6 and Fig. 7 show that the proposed algorithm's FTP and HTTP throughput is less than the PF algorithm. Because the priority of these services is less than guaranteed bit rate services and resources firstly will be allocated to these services. On the other hand, instead of service non-guaranteed bit rate users with good channel quality, some resources are assigned to the users which have poor channel quality for transmission guaranteed bit rates data; it tends to reduce its throughput.



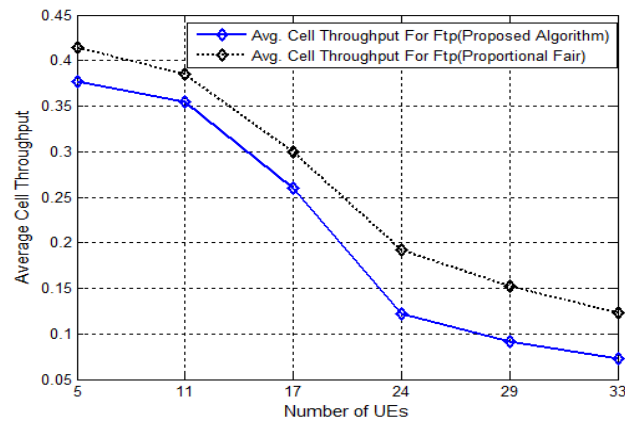


Fig. 6. Comparison of the FTP throughput.

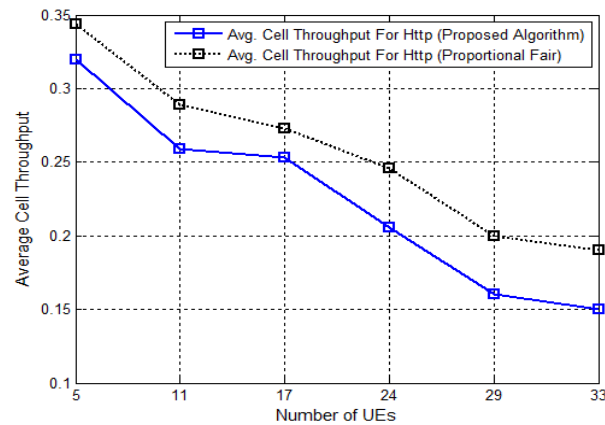


Fig. 7. Comparison of the HTTP throughput.

Fig. 8 and Fig. 9 respectively show, fairness and delay of VOIP and Video services (as representative of real-time service) and HTTP service (as representative of non-real-time service). In these figures with the increasing number of users, VOIP and Video services have better fairness and lower latency in comparison to the PF algorithm. Of course, this improvement is associated with eliminating some performance of HTTP service; this is because of the priority of guaranteed bit rate services to the non-guaranteed bit rate services. This leads us to our goal to ensure the QoS of real-time service, at the cost of destroying a certain amount of non-real-time services performance.

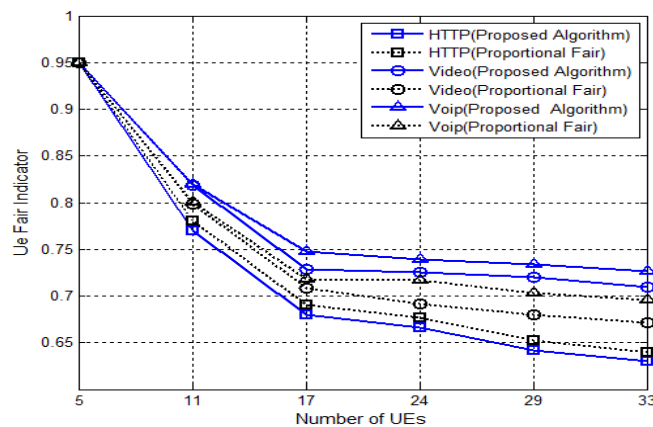


Fig. 8. Comparison of the Fairness of the service.

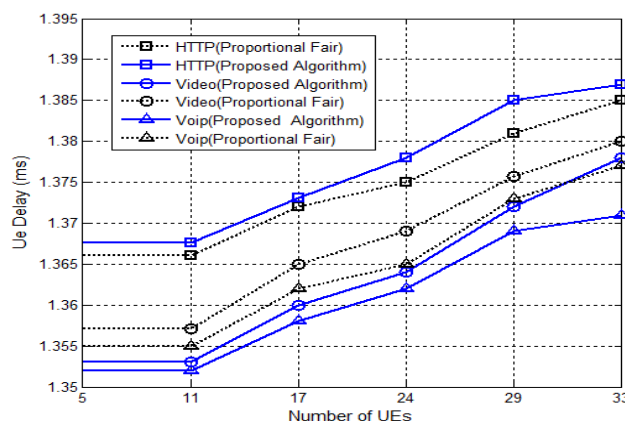


Fig. 9. Comparison of the delay of the service.

## 6 | Conclusion

In this paper, users with five different services (VOIP, Video, Gaming, FTP, and HTTP) were defined to show how the proposed scheduling algorithm manages and services mixed traffic of various services. Some of them are serviced with a higher priority due to their real-time nature. To evaluate the performance of the proposed algorithm for multimedia applications, the number of users was increased and the service quality at both system and user levels was measured. For N users covered by eNodeB, QoS of different classes was evaluated. Also, the quality of service between the proposed algorithm and PF algorithm was compared and average throughput for different traffic classes was evaluated. The simulation results showed that the proposed scheduling algorithm maximized system throughput also with the prioritization of services and taking into account the user's buffer status minimized packet delay. Also, taking into account the user channel conditions, maintained fairness between them. This algorithm can increase the function of real-time services by destroying a certain amount of non-real-time service performance. Also, it can be concluded, when the level of QoS is required, it is important to be aware of real-time traffic by the scheduling algorithm and if multimedia traffic doesn't have priority, it leads to a service drop for real-time services.

## References

- [1] Dahlman, E., Parkvall, S., Sköld, J., & Beming, P. (2007). *3G evolution: HSPA and LTE for mobile broadband*. Academic Press.
- [2] Khan, F. (2009). *LTE for 4G mobile broadband: air interface technologies and performance* (Vol. 9780521882). Cambridge University Press.
- [3] Holma, H., & Toskala, A. (2009). *LTE for UMTS - OFDMA and SC-FDMA based radio access*. John Wiley & Sons.
- [4] Sesia, S., Toufik, I., & Baker, M. (2009). *LTE-the UMTS long term evolution*. Wiley Online Library.
- [5] Trivedi, R. D., & Patel, M. C. (2014). Comparison of different scheduling algorithm for LTE. *International journal of emerging technology and advanced engineering*, 4(5), 334–339.
- [6] Iosif, O., & Bănică, I. (2013). Performance analysis of Downlink LTE using system level simulator. *UPB sci. bull., series C*, 75(1), 111–122. [http://scientificbulletin.upb.ro/rev\\_docs\\_arhiva/full58a\\_785038.pdf](http://scientificbulletin.upb.ro/rev_docs_arhiva/full58a_785038.pdf)
- [7] Beh, K. C., Armour, S., & Doufexi, A. (2008). Joint time-frequency domain proportional fair scheduler with harq for 3gpp lte systems. *2008 IEEE 68th vehicular technology conference* (pp. 1–5). IEEE.
- [8] Badia, L., Baiocchi, A., Todini, A., Merlin, S., Pupolin, S., Zanella, A., & Zorzi, M. (2007). On the impact of physical layer awareness on scheduling and resource allocation in broadband multicellular IEEE 802.16 systems. *IEEE wireless communications*, 14(1), 36–43. DOI:10.1109/MWC.2007.314549
- [9] Chadchan, S. M., & Akki, C. B. (2013). A fair downlink scheduling algorithm for 3GPP LTE networks. *International journal of computer network and information security*, 5(6), 34–41.
- [10] Elhadad, M. I., & Abd-elnaby, M. (2014). Enhanced PF scheduling algorithm for LTE downlink system. *Mobile computing journal*, 3(1), 7–12. [https://www.academia.edu/download/48138766/MC10045\\_3\\_0\\_7\\_12.pdf](https://www.academia.edu/download/48138766/MC10045_3_0_7_12.pdf)

- [11] Sharma, V., & Chopra, P. P. K. (2014). A packet based scheduling mechanism in real time traffic for LTE downlink networks. *International journal of engineering research & technology*, 3(9), 165–169. <https://www.academia.edu/download/64322866/a-packet-based-scheduling-mechanism-in-real-IJERTV3IS090167.pdf>
- [12] Müller, M. K., Schwarz, S., & Rupp, M. (2013). QoS investigation of proportional fair scheduling in lte networks. *2013 IFIP wireless days (WD)* (pp. 1–4). IEEE. <https://ieeexplore.ieee.org/abstract/document/6686478>
- [13] Wang, Y. H., & Huang, H. Y. (2014). A qos-based fairness-aware downlink scheduling in lte-advanced. *2014 17th international conference on network-based information systems* (pp. 470–475). IEEE. <https://ieeexplore.ieee.org/abstract/document/7023996>
- [14] Ma, P., Lu, Y., Hou, Y., Li, L., Zhao, X., Zhang, L., & Zhu, H. (2018). A multi-service qos guaranteed scheduling algorithm for TD-LTE 230 MHz power wireless private networks. *2018 12th international symposium on antennas, propagation and em theory (ISAPE)* (pp. 1–4). IEEE. <https://ieeexplore.ieee.org/abstract/document/8634219>
- [15] Asadollahi, F., & Dehdasht-Heydari, R. (2018). Introduction of a novel hybrid weighted exponential logarithm-maximum throughput (HWEL-MT) scheduler for QoS improvement of LTE/4G cellular networks. *Wireless personal communications*, 98(1), 91–104. DOI:10.1007/s11277-017-4857-0
- [16] Costa Neto, F. H., Bezerra Rodrigues, E., Aguiar Sousa, D., Ferreira Maciel, T., & Porto Cavalcanti, F. R. (2017). QoS-aware scheduling algorithms to enhance user satisfaction in OFDMA systems. *Transactions on emerging telecommunications technologies*, 28(10), e3165. DOI:10.1002/ett.3165
- [17] Chaudhuri, S., Baig, I., & Das, D. (2018). A novel QoS aware medium access control scheduler for LTE-advanced network. *Computer networks*, 135, 1–14. DOI:10.1016/j.comnet.2018.01.024
- [18] Uyan, O. G., & Gungor, V. C. (2019). QoS-aware LTE-a downlink scheduling algorithm: a case study on edge users. *International journal of communication systems*, 32(15), e4066. DOI:10.1002/dac.4066
- [19] Madi, N. K. M., Hanapi, Z. M., Othman, M., & Subramaniam, S. K. (2018). Delay-based and QoS-aware packet scheduling for RT and NRT multimedia services in LTE downlink systems. *Eurasip journal on wireless communications and networking*, 2018(1), 1–21. DOI:10.1186/s13638-018-1185-3
- [20] Rocha, F. G. C., & Vieira, F. H. T. (2019). A channel and queue-aware scheduling for the LTE downlink based on service curve and buffer overflow probability. *IEEE wireless communications letters*, 8(3), 729–732. DOI:10.1109/LWC.2018.2889725
- [21] Elhadad, M. I., El-Shafai, W., El-Rabaie, E. S. M., Abd-Elnaby, M., & Abd El-Samie, F. E. (2020). Enhanced fair earliest due date first scheduling strategy for multimedia applications in LTE downlink framework. *International journal of communication systems*, 33(6), e4190. DOI:10.1002/dac.4190
- [22] Nasralla, M. M. (2020). A hybrid downlink scheduling approach for multi-traffic classes in LTE wireless systems. *IEEE access*, 8, 82173–82186. DOI:10.1109/ACCESS.2020.2990381
- [23] Ashfaq, K., Safdar, G. A., & Ur-Rehman, M. (2021). Comparative analysis of scheduling algorithms for radio resource allocation in future communication networks. *PeerJ computer science*, 7, 1–13. DOI:10.7717/PEERJ-CS.546
- [24] Mushtaq, M. S., Shahid, A., & Fowler, S. (2012). QoS-aware lte downlink scheduler for voip with power saving. *2012 IEEE 15th international conference on computational science and engineering* (pp. 243–250). IEEE. <https://ieeexplore.ieee.org/abstract/document/6417300>
- [25] Stolyar, A. L., & Ramanan, K. (2001). Largest weighted delay first scheduling: large deviations and optimality. *Annals of applied probability*, 11(1), 1–48. DOI:10.1214/aoap/998926986
- [26] Dong, S. (2006). Methods for constrained optimization. *Massachusetts institute of technology, massachusetts*. [https://www.researchgate.net/profile/Shuonan-Dong-2/publication/255602767\\_Methods\\_for\\_Constrained\\_Optimization/links/00b7d53c5c41574549000000/Methods-for-Constrained-Optimization.pdf](https://www.researchgate.net/profile/Shuonan-Dong-2/publication/255602767_Methods_for_Constrained_Optimization/links/00b7d53c5c41574549000000/Methods-for-Constrained-Optimization.pdf)
- [27] Huang, S. F., Wu, E. H. K., & Chang, P. C. (2006). *Optimal adaptive voice smoother with lagrangian multiplier method for VoIP service* [presentation]. Proceedings of the 6th WSEAS international conference on multimedia systems & signal processing (pp. 186–191). <https://dl.acm.org/doi/abs/10.5555/1974030.1974070>
- [28] Zöchmann, E., Schwarz, S., Pratschner, S., Nagel, L., Lerch, M., & Rupp, M. (2016). Exploring the physical layer frontiers of cellular uplink: the vienna LTE-a uplink simulator. *Eurasip journal on wireless communications and networking*, 2016(1), 1–18. DOI:10.1186/s13638-016-0609-1
- [29] Jain, R. (1990). *The art of computer systems performance analysis techniques for experimental design, measurement, simulation and modeling* (Vol. 1). New York: Wiley.